AMENDMENTS TO THE SPECIFICATION

Please amend the paragraph beginning at page 21 line 31 as follows.

Figure 5 is block diagram of a stereo image correction system 122, which inputs the left and right stereo signals 126 and 128. The image-correction system 122 corrects the distorted spectral densities of various sound systems by advantageously dividing the audible frequency spectrum into a first frequency component, containing relatively lower frequencies, and a second frequency component, containing relatively higher frequencies. Each of the left and right signals 126 and 128 is separately processed through corresponding low-frequency correction systems 580, 582, and high-frequency correction systems 584 and 586. It should be pointed out that in one embodiment the correction systems 580 and 582 will operate in a relatively "low" frequency range of approximately 100 to 1000 Hertz, while the correction systems 584 and 586 will operate in a relatively "high" frequency range of approximately 1000 to 10,000 Hertz. This is not to be confused with the general audio terminology wherein low frequencies represent frequencies up to 100 Hertz, mid frequencies represent frequencies between 100 Hz to 4 kHz, and high frequencies represent frequencies above 4 kHz.

Please amend the paragraph beginning at page 23 line 27 as follows.

The shaped difference signal is provided to a mixer 542, which also receives the sum signal from the device ~~506~~508. In one embodiment, the stereo signals 594 and 596 are also provided to the mixer 542. All of these signals are combined within the mixer 542 to produce an enhanced and spatially-corrected left output signal 530 and right output signal 532.

Please amend the paragraphs beginning at page 24 line 24 as follows.

As can be seen in Figures 6A-6C, spatial correction of an audio signal by the systems 580, 582, 584, and 586 is substantially uniform within the pass-bands, but is largely frequency-dependent within the transition bands. The amount of acoustic correction applied to an audio signal can be varied as a function of frequency through adjustment of the stereo image correction system ~~622~~ 122 which varies the slope of the transition bands of Figures 6A-6C. As a result, frequency-dependent correction is applied to a first frequency range between 100 and 1000 hertz, and applied to a second

frequency range of 1000 to 10,000 hertz. An infinite number of correction curves are possible through independent adjustment of the correction systems 580, 582, 584 and 586.

In accordance with one embodiment, spatial correction of the higher frequency stereo-signal components occurs between approximately 1000 Hz and 10,000 Hz. Energy correction of these signal components may be positive, i.e., boosted, as depicted in Figure 6B, or negative, i.e., attenuated, as depicted in Figure 6C. The range of boost provided by the correction systems 584, 586 is characterized by a maximum-boost curve 660 and a minimum-boost curve ~~162~~662. Curves 664, 666, and 668 represent still other levels of boost, which may be required to spatially correct sound emanating from different sound reproduction systems. Figure 6C depicts energy-correction curves that are essentially the inverse of those in Figure 6B.

Since the lower frequency and higher frequency correction factors, represented by the curves of Figures 6A-6C, are added together, there is a wide range of possible spatial correction curves applicable between the frequencies of 100 to 10,000 Hz. Figure 6D is a graphical representation depicting a range of composite spatial correction characteristics provided by the stereo image correction system ~~522~~122. Specifically, the solid line curve 680 represents a maximum level of spatial correction comprised of the curve 650 (shown in Fig. 6A) and the curve 660 (shown in Fig. 6B). Correction of the lower frequencies may vary from the solid curve 680 through the range designated by θ_1. Similarly, correction of the higher frequencies may vary from the solid curve 680 through the range designated by θ_2. Accordingly, the amount of boost applied to the first frequency range of 100 to 1000 Hertz varies between approximately 0 and 15 dB, while the correction applied to the second frequency range of 1000 to 10,000 Hertz may vary from approximately 13 dB to -15 dB.

Please amend the paragraph beginning at page 25 line 24 as follows.

Turning now to the stereo image enhancement aspect of the present invention, a series of perspective-enhancement, or normalization curves, is graphically represented in Figure 7. The signal $(L_c\text{-}R_c)_p$ ~~in equations 1 and 2~~ above represents the processed difference signal which has been spectrally shaped according to the frequency-response

characteristics of Figure 7. These frequency-response characteristics are applied by the equalizer 520 depicted in Figure 5 and are partially based upon HRTF principles.

Please amend the paragraph beginning at page 26 line 14 as follows.

According to one embodiment, the range for the perspective curves of Figure 7 is defined by a maximum gain of approximately 10-15 dB located at approximately 125 to 150 Hz. The maximum gain values denote a turning point for the curves of Figure 7 whereby the slopes of the curves 790, 792, 794, 796, and 798 change from a positive value to a negative value. Such turning points are labeled as points A, B, C, D, and E in Figure 7. The gain of the perspective curves decreases below 125 Hz at a rate of approximately 6 dB per octave. Above 125 Hz, the gain of the curves of Figure 7 also decreases, but at variable rates, towards a minimum-gain turning point of approximately -2 to +10 dB. The minimum-gain turning points vary significantly between the curves 790, 792, 794, 796, and 798. The minimum-gain turning points are labeled as points A', B', C', D', and E', respectively. The frequencies at which the minimum-gain turning points occur varies from approximately 2.1 kHz for curve 790 to approximately 5 kHz for curve 798. The gain of the curves 790, 792, 794, 796, and 798 increases above their respective minimum-gain frequencies up to approximately 10 ~~Khz~~kHz. Above 10 ~~Khz~~kHz, the gain applied by the perspective curves begins to level off. An increase in gain will continue to be applied by all of the curves, however, up to approximately ~~120~~ 20 ~~Khz~~kHz, i.e., approximately the highest frequency audible to the human ear.

Please amend the paragraph beginning at page 31 line 11 as follows.

For example, a person listening to the acoustic energy represented by the spectral lines 1004 and 1002 will perceive acoustic energy at 50 Hz, as shown by the spectral line 1006, at 60 Hz, as shown by the spectral line ~~1006~~1008, and at 110 Hz, as shown by the spectra line 1010. The spectral line 1010 does not correspond to real acoustic energy produced by the loudspeaker, but rather corresponds to a spectral line created inside the ear by the non-linearities of the ear. The line 1010 occurs at a frequency of 110 Hz which is the sum of the two actual spectral lines (110 Hz = 50 Hz + 60 Hz). Note that the non-linearities of the ear will also create a spectral line at the difference frequency of 10 Hz (

10 Hz = 60 Hz - 50 Hz), but that line is not perceived because it is below the range of human hearing.

Please amend the paragraph beginning at page 33 line 22 as follows.

Figure 13B is a block diagram of a topology for a two-channel bass enhancement unit 1304 having a first input 1309, a second input 1311, a first output 1317 and a second output 1319. The first input 1309 and first output 1317 correspond to a first channel. The second input 1311 and second output 1319 correspond to a second channel. The first input 1309 is provided to a first input of a combiner 1310 and to an input of a signal processing block 1313. An output of the signal processing block 1313 is provided to a first input of a combiner 1314. The second input 1311 is provided to a second input of the combiner 1310 and to an input of a signal processing block 1315. An output of the signal processing block ~~1615~~ 1315 is provided to a first input of a combiner 1316. An output of the combiner 1310 is provided to an input of a signal processing block 1312. An output of the signal processing block 1312 is provided to a second input of the combiner 1314 and to a second input of the combiner 1316. An output of the combiner 1314 is provided to the first output 1317. An output of the second combiner 1316 is provided to the second output 1319.

Please amend the paragraph beginning at page 35 line 3 as follows.

The output of the adder 1406 is provided to ~~an input of a lowpass filter 1409. An output of the lowpass filter 1409 is provided to a~~ first bandpass filter 1412, a second bandpass filter 1413, a third bandpass filter 1415, and a fourth bandpass filter 1411 ~~and the fifth bandpass filter 1414~~. The output of the bandpass filter 1413 is provided to an input of an adder 1418.

Please amend the paragraphs beginning at page 35 line 12 as follows.

The output of the bandpass filter 1412 is provided to ~~a first throw of a single pole double throw (SPDT) switch 1419. The output of the bandpass filter 1414 is provided to a second throw of the SPDT switch 1419. The pole of the switch 1419 is provided to~~ an input of the adder 1418.

An output of the adder 1418 is provided to an input of the bass punch unit 1420. An output of the bass punch unit 1420 is provided to a first throw of a (SPDT) switch 1422. A second throw of the SPDT switch 1422 is provided to ground. ~~The throw~~A pole of the SPDT switch 1422 is provided to a first input of a left-channel adder 1424 and to a first input of a right-channel adder 1432. The left-channel input 1402 is provided to a second input of the left-channel adder 1424 and the right-channel input 1404 is provided to a second input of the right-channel adder 1432. The outputs of the left-channel adder 1424 and the right-channel adder 1432 are, respectively, a left-channel output 1430 and a right-channel output 1433 of the signal processing block 1400. The switches 1422 and 1416 are optional and may be replaced by fixed connections.

The switches 1416 ~~and 1419~~ allows the filters 1411-1415 to be configured for ~~three~~ two different frequency ranges, namely 40-~~100, 60~~-150 Hz, and 100-200 Hz.

The filtering operations provided by the filters 1411-1413, 1415 and the combiner 1418 may be combined into a composite filter 1407 as shown in Figure 14. For example, in an alternative embodiment, the filters 1411-1413, 1415 are combined into a single bandpass filter having a passband that extends from approximately 40 Hz to 250 Hz. For processing bass frequencies, the passband of the composite filter 1407 preferably extends from approximately 20 to 100 Hz at the low-end, and from approximately 150 to 350 Hz at the high-end. The composite filter 1407 may have other filter transfer functions as well, including, for example, a highpass filter, a shelving filter, etc. The composite filter may also be configured to operate in a manner similar to a graphic equalizer and attenuate some frequencies within its passband relative to other frequencies within its passband.

As shown, Figure 14 corresponds approximately to the topology shown in Figure 13B, where the signal processing blocks 1313 and 1315 have a transfer function of unity and the signal processing block 1312 comprises the composite filter 1407 and the bass punch unit 1420. However, the signal processing shown in Figure 14 is not limited to the topology shown in Figure 13B. The elements of Figure 14 may also be used in the topology shown in Figure 13C, where the signal processing blocks 1321 and 1323 have a transfer function of unity and the signal processing blocks 1322 and 1324 comprise the composite filter 1407 and the bass punch unit 1420. Although not shown in Figure 14, the signal processing blocks 1313, 1315, 1321, and 1323 may provide additional signal

processing, such as, for example, high pass filtering to remove low bass frequencies, high pass filtering to remove frequencies processed by the bass punch unit ~~1402~~1420, high frequency emphasis to enhance the high frequency sounds, additional mid bass processing to supplement the bass punch system, etc. Other combinations are contemplated as well.

Figure 15 is a frequency-domain plot that shows the general shape of the transfer functions of the bandpass filters 1411-1413, 1415. Figure 15 shows the bandpass transfer functions 1501-~~1505~~1504, corresponding to the bandpass filters 1411-1413, 1415 respectively. The transfer functions 1501-~~1505~~ 1504 are shown as bandpass functions centered at 40, ~~60,~~ 100, 150, and 200 Hz respectively.

In one embodiment, the bandpass filter 1411 is tuned to a frequency below 100 Hz, such as 40 Hz. When the switch 1416 is in a first position, corresponding to the first throw, it selects the bandpass filter 1411 and deselects the bandpass filter 1415, thereby providing bandpass filters at 40, ~~60, and~~ 100, and 150 Hz. When the switch 1416 is in a second position, corresponding to the second throw, it deselects the bandpass filter 1411 and selects the bandpass filter 1415, thus providing bandpass filters at~~, 60,~~ 100, ~~and~~ 150, and 200 Hz.

Thus, the switch 1416 desirably allows a user to select the frequency range to be enhanced. A user with a loudspeaker system that provides small woofers, such as woofer of three to four inches in diameter, will typically select the upper frequency range provided by the bandpass filters 1412-1413, 1415 which are tuned to ~~40, 60,~~ 100, ~~and~~ 150, and 200 Hz respectively. A user with a loudspeaker system that provides somewhat larger woofers, such as woofers of approximately five inches in diameter or larger, will typically select the lower frequency range provided by the bandpass filters 1411-1413, ~~1515~~ which are tuned to 40, ~~60,~~ 100, and 150 Hz respectively. One skilled in the art will recognize that more switches could be provided to allow selection of more bandpass filters and more frequency ranges. Selecting different bandpass filters to provide different frequency ranges is a desirable technique because the bandpass filters are inexpensive and because different bandpass filters can be selected with a single-throw switch.

Please amend the paragraph beginning at page 37 line 24 as follows.

In response to an increase in the amplitude of the envelope of the signal provided to the input of the bass punch unit 1420, the servo loop increases the forward gain of the bass punch unit 1420. Conversely, in response to a decrease in the amplitude of the envelope of the signal provided to the input of the bass punch unit 1420, the servo loop ~~increases~~ decreases the forward gain of the bass punch unit 1420. In one embodiment, the gain of the bass punch unit 1420 increases more rapidly that the gain decreases. Figure 16 is a time domain plot that illustrates the gain of the bass punch unit 1420 in response to a unit step input. One skilled in the art will recognize that Figure 16 is a plot of gain as a function of time, rather than an output signal as a function of time. Most amplifiers have a gain that is fixed, so gain is rarely plotted. However, the Automatic Gain Control (AGC) in the bass punch unit 1420 varies the gain of the bass punch unit 1420 in response to the envelope of the input signal.

Please amend the paragraph beginning at page 38 line 11 as follows.

The attack time constant 1604 and the decay time constant 1606 are desirably selected to provide enhancement of the bass frequencies without overdriving other components of the system such as the amplifier and loudspeakers. Figure 17 is a time-domain plot 1700 of a typical bass note played by a musical instrument such as a bass guitar, bass drum, synthesizer, etc. The plot 1700 shows a higher-frequency portion ~~1740~~ 1744 that is amplitude modulated by a lower-frequency portion having a modulation envelope 1742. The envelope 1742 has an attack portion 1746, followed by a decay portion 1747, followed by a sustain portion 1748, and finally, followed by a release portion 1749. The largest amplitude of the plot 1700 is at a peak 1750, which occurs at the point in time between the attack portion 1746 and the decay portion 1747.

Please amend the paragraph beginning at page 39 line 4 as follows.

Similarly, a drumhead, when struck, will produce an initial set of large excursion vibrations corresponding to the attack portion 1746 and the decay portion 1747. After the large excursion vibrations have died down (corresponding to the end of the decay portion ~~1717~~1747) the drumhead will continue to vibrate for a period of time corresponding to the sustain portion 1748 and release portion 1749. Many musical instrument sounds can be created merely by controlling the length of the periods 1746-~~2049~~1749.

Please amend the paragraphs beginning at page 40 line 22 as follows.

Figure 18 is a signal processing block diagram of a bass enhancement system 1800 that provides bass enhancement using a peak compressor to control the amplitude of pulses, such as the initial pulse, bass notes. In the system 1800, a peak compressor 1802 is interposed between the combiner ~~1418~~ 1718 and the punch unit ~~1420~~1720. The output of the combiner ~~1418~~ 1718 is provided to an input of the peak compressor 1802, and an output of the peak compressor 1802 is provided to the input of the bass punch unit ~~1420~~1720.

The comments above relating Figure 14 to Figures 13B and 13C apply to the topology shown in Figure 18 as well. For example, as shown, Figure 18 corresponds approximately to the topology shown in Figure 13B, where the signal processing blocks 1313 and 1315 have a transfer function of unity and the signal processing block 1312 comprises the composite filter ~~1407~~1707, the peak compressor 1802, and the bass punch unit ~~1420~~1720. However, the signal processing shown in Figure 18 is not limited to the topology shown in Figure 13B. The elements of Figure 18 may also be used in the topology shown in Figure 13C. Although not shown in Figure 18, the signal processing blocks 1313, 1315, 1321, and 1323 may provide additional signal processing, such as, for example, high pass filtering to remove low bass frequencies, high pass filtering to remove frequencies processed by the bass punch unit ~~1402~~ 1720 and the compressor 1802, high frequency emphasis to enhance the high frequency sounds, additional mid bass processing to supplement the bass punch system ~~1420~~ 1720 and peak compressor 1802, etc. Other combinations are contemplated as well.

Please amend the paragraph beginning at page 41 line 19 as follows.

Figure 19 is a time-domain plot showing the effect of the peak compressor on an envelope with an initial pulse of relatively high amplitude. –Figure 19 shows a time-domain plot of an input envelope 1914 having an initial large amplitude pulse followed by a longer period of lower amplitude signal. An output envelope 1916 shows the effect of the bass punch unit ~~1420~~ 1720 on the input envelope 1914 (without the peak compressor 1802). An output envelope 1917 shows the effect of passing the input signal 1914 through both the peak compressor 1802 and the punch unit ~~1420~~1720.

Please amend the paragraph beginning at page 45 line 8 as follows.

Figure 21 is a block diagram of a system that uses differential amplifiers to provide the differential perspective correction shown in Figure 20. In Figure 21, the first input 2010 is provided to a non-inverting input of a first differential amplifier 2102 and to a first input of a cross-over impedance block 2106. The second input 2012 is provided to a non-inverting input of a second differential amplifier 2104 and to a second terminal of the cross-over impedance block 2106. An ~~non-~~inverting input of the first differential amplifier 2102 is provided to a first terminal of a cross-over impedance block 2107 and to a first terminal of a first feedback impedance 2108. An output of the first differential amplifier 2102 is provided to the first output 2030 and to a second terminal of the first feedback impedance 2108. An ~~non-~~inverting input of the second differential amplifier 2104 is provided to a second terminal of the cross-over impedance block 2107 and to a first terminal of a second feedback impedance ~~2108~~2109. An output of the second differential amplifier 2104 is provided to the second output 2032 and to a second terminal of the second feedback impedance 2109.

Please amend the paragraphs beginning at page 46 line 7 as follows.

With such a reference, the overall correction curve 2300 is defined by two turning points labeled as point A and point B. At point A, which in one embodiment is approximately ~~2125~~ 125 Hz, the slope of the correction curve changes from a positive value to a negative value. At point B, which in one embodiment is approximately ~~21.8~~2 kHz, the slope of the correction curve changes from a negative value to a positive value.

Thus, the frequencies below approximately ~~2125~~ 125 Hz are de-emphasized relative to the frequencies near ~~2125~~ 125 Hz. In particular, below ~~2125~~ 125 Hz, the gain of the overall correction curve 2300 decreases at a rate of approximately 6 dB per octave. This de-emphasis of signal frequencies below ~~2125~~ 125 Hz prevents the over-emphasis of very low, (i.e. bass) frequencies. With many audio reproduction systems, over emphasizing audio signals in this low-frequency range relative to the higher frequencies can create an unpleasurable and unrealistic sound image having too much bass response. Furthermore, over emphasizing these frequencies may damage a variety of audio components including the loudspeakers.

Between point A and point B, the slope of one overall correction curve is negative. That is, the frequencies between approximately ~~2125~~ 125 Hz and approximately ~~21.8~~2 kHz are de-emphasized relative to the frequencies near ~~2125~~ 125 Hz. Thus, the gain associated with the frequencies between point A and point B decrease at variable rates towards the maximum-equalization point of -8 dB at approximately ~~21.8~~2 kHz.

Above ~~21.8~~2 kHz the gain increases, at variable rates, up to approximately ~~120~~ 20 kHz, i.e., approximately the highest frequency audible to the human ear. That is, the frequencies above approximately ~~21.8~~2 kHz are emphasized relative to the frequencies near ~~21.8~~2 kHz. Thus, the gain associated with the frequencies above point B increases at variable rates towards ~~120~~ 20 kHz.

Please amend the paragraph beginning at page 48 line 1 as follows.

Examples of HRTF transfer functions which can be used to achieve a certain perceived azimuth are described in the article by E.A.B. Shaw entitled "Transformation of Sound Pressure Level From the Free Field to the Eardrum in the Horizontal Plane", J.Acoust.Soc.Am., Vol. 106, No. 6, December 1974, and in the article by S. Mehrgardt and V. Mellert entitled "Transformation Characteristics of the External Human Ear", J.Acoust.Soc.Am., Vol. 61, No. 6, June 1977~~, both of which are incorporated herein by reference as though fully set forth~~.

Please amend the paragraph beginning at page 48 line 9 as follows.

Figure 24 is a block diagram of one embodiment of a sound enhancement system 2400 that can be implemented on a single chip. As described in connection with Figures 1-23 above, the system 2400 includes a vertical image enhancement block 2402, a bass enhancement block 2404 and a ~~vertical~~ horizontal image enhancement block 2406. External connections to the system 2400 are provided through connector pins P1-P27. A positive supply voltage is provided to the pin P25, a negative supply voltage is provided to the pin P26, and a ground is provided to the pin P27. A first terminal of a compression coupling capacitor 2421 is provided to the pin P10 and a second terminal of the compression coupling capacitor 2421 is provided to the pin P11. A first terminal of a compression delay capacitor 2420 is provided to the pin P13 and a second terminal of the compression delay capacitor 2420 is provided to the pin P14. A first terminal of a width-

control resistor 2430 is provided to the pin P19 and a second terminal of the width-control resistor 2430 is provided to the pin P20. A first terminal of a width-control resistor 2431 is provided to the pin P21 and a second terminal of the width-control resistor 2431 is provided to the pin P22. In one embodiment, the width-control resistors 2430 and 2431 are variable resistors.

Please amend the paragraph beginning at page 49 line 16 as follows.

The ~~left~~right-channel shown in Figure 25B is similar to the ~~right~~ left channel shown in Figure 25A, having a bypass input from the pin P3, a right-channel input from the pin P4 and a right-channel output 2514.

Please amend the paragraph beginning at page 51 line 7 as follows.

The pin P6 is provided to a first terminal of a capacitor 2724 and to a first terminal of a capacitor 2728. A second terminal of the capacitor 2728 is provided to a first terminal of a resistors 2725, to a first terminal of a resistor 2726, and to an inverting input of an opamp 2729. A non-inverting input of the opamp 2729 is provided to ground. An output of the opamp 2729 is provided to a second terminal of the capacitor 2724, to a second terminal of the resistor 2725, to a second terminal of the resistors 2726, and to a first terminal of a resistor 2730. The second terminal of the capacitor 2724 is provided to the pin P8. A second terminal of the resistor 2725 is provided to the pin P9. A second terminal of the resistor 2730 is provided to the second filter output.

Please amend the paragraph beginning at page 59 line 3 as follows.

Referring now to Figure ~~27~~33, the third crossover network 3074 interconnects the collectors of transistors 3010 and 3012. The third crossover network 3074 includes the resistor 3064 and the capacitor 3028 which are selected to create a low-pass filter which de-emphasizes frequencies above a mid-range of frequencies. In one embodiment, the cut-off frequency of the low-pass filter is approximately 795 Hz. Preferably, the value of resistor 3064 is approximately 9.09 kohm and the value of the capacitor 3028 is approximately 0.022 microfarads.

Please amend the paragraph beginning at page 59 line 25 as follows.

Thus, the frequencies below approximately 125 Hz are de-emphasized relative to the frequencies near 125 Hz. In particular, below 125 Hz, the gain of the overall correction curve ~~800~~ 2300 decreases at a rate of approximately 6 dB per octave. This de-emphasis of signal frequencies below 125 Hz prevents the over-emphasis of very low, (i.e., bass) frequencies. With many audio reproduction systems, over emphasizing audio signals in this low-frequency range relative to the higher frequencies can create an unpleasurable and unrealistic sound image having too much bass response. Furthermore, over emphasizing these frequencies may damage a variety of audio components, including the loudspeakers.

Please amend the paragraph beginning at page 63 line 9 as follows.

The above embodiments of a differential perspective correction apparatus can also include output buffers 3630 as illustrated in Figure 36. The output buffers ~~3600~~ 3630 are designed to isolate the perspective correction differential apparatus from variations in the load presented by a circuit connected to the left and right output terminals 3004 and 3006. For example, when the left and right output terminals 3004 and 3006 are connected to a pair of loudspeakers, the impedance load of the loudspeakers will not alter the manner in which the differential perspective correction apparatus equalizes the differential signal. Accordingly, without the output buffers 3630, circuits, loudspeakers and other components will affect the manner in which the differential perspective correction apparatus 102 equalizes the differential signal.

Please amend the paragraphs beginning at page 64 line 12 as follows.

Figure 37 shows yet another embodiment of the stereo image enhancement processor 124. In Figure 37, the left input 2630 is provided to a first terminal of a resistor 3710, to a first terminal of a resistor 3716, and to a first terminal of a resistor 3740. The second terminal of the resistor 3710 is ~~to~~ provided to a first terminal of a resistor 3711, and to an ~~non~~-inverting input of an opamp 3712. The right input 2631 is provided to a first terminal of a resistor 3713, to a first terminal of a resistor 3741, and to a first terminal of a resistor 3746. The second terminal of the resistor 3713 is provided to a first terminal of the resistor 3714 and to a non-inverting input of the opamp 3712. The second terminal of the resistor 3714 is provided to ground. The second terminal of

the resistor 3740 and a second terminal of the resistor 3741 are provided to a non-inverting input of the opamp 3744, and to a first terminal of the resistor 3742. The second terminal of the resistor 3742 provided to ground.

The output of the opamp 3744 in provided a first terminal of the resistor 3761. A second terminal of the resistor 3761 is provided to an inverting input of the opamp 3744. The second terminal of the resistor 3743 is provided to ground. Returning to the opamp 3712, an output of the opamp 3712 is provided to a second terminal of the resistor 3711. The output of the opamp 3712 is also provided in first terminal of the resistor 3715. The second terminal of the resistor 3715 provided to a first terminal of a capacitor 3717. -A second terminal of the capacitor 3717 is provided to a first terminal of the resistor 3718, to a first terminal of the resistor 3719, to a first terminal of a capacitor 3721, and to a first terminal of a resistor 3722. The second terminal of the resist-or 3718 provided ground. The second terminal of the resistor 3719 provided to a second terminal of the resistor 3720, and to the second terminal of the resistor 3725. The second terminal of the ~~resistor~~ capacitor 3721 is provided to a first terminal of the resistor 3720 and to a first terminal of -the resistor ~~3723~~3023. The second terminal of the resistor 3722 is provided to a first terminal of the resistor 3725 and to a first terminal of a capacitor 3724. The second terminal of the resistor ~~3723~~ 3023 and ~~he~~ the second terminal of the capacitor ~~3724~~ 3024 are both provided to ground.

The second terminal of the resistor 3719 is also provided to a first terminal of a resistor 3726 and to an inverting input of an opamp 3727. A non-inverting input of the opamp 3727 is provided to ground. The second terminal of the resistor 3726 is provided to an output of the opamp 3727. The output of the opamp 3727 is provided to a first fixed terminal of a potentiometer 3728. A second fixed terminal of the potentiometer 3728 is provided ground. A wiper of the potentiometer 3728 is provided to the ~~first~~ second terminal of a resistor 3747 and to a first terminal of a resistor ~~3720~~3729.

Please amend the paragraph beginning at page 65 line 21 as follows.

A non-inverting input of the opamp is 3749 provided to ground. An output of the opamp 3749 is provided to second terminal of the resistor 3748 and to the first terminal of the resistor 3750. The second terminal of the resistor 3750 is provided to a second

terminal of the resistor 3729. A second terminal of the resistor 3730 provided to a non-inverting input of the opamp 3753. A first terminal of the resistor 3731 is also provided to the non-inverting input of the opamp 3735. The second terminal of the resistor 3731 is provided to ground. An ~~non-~~inverting input of the opamp 3735 is provided to a first terminal of a resistor 3734 and to a first terminal of a resistor 3732. The second terminal of the resistor 3732 provided to ground. An output of the opamp 3735 provided to a second terminal of a resistor 3734. A second terminal of the resistor 3750, a second terminal of the resistor 3751, a second terminal of the resistor 3746, and a first terminal of a resistor 3752 are all provided to a non-inverting input of an opamp 3755. A second terminal of the resistor 3752 is provided to ground. A non-inverted input of the opamp 3755 is provided to a first terminal of a resistor 3753 and to a first terminal of a resistor 3754. An output of the opamp 3755 is provided to a second terminal of the resistor 3754.

Please amend the paragraph beginning at page 67 line 24 as follows.

Output of the summer ~~3838~~ 3830 provided an input of a 5 ~~db~~dB attenuator 3832. An output the attenuator 3832 provided to first input of a summer 3835 and to a first input of a summer 3866. An output of the attenuator 3833 is provided to a second input of the summer 3835. An output of the attenuator 3834 is provided to a second input of the summer 3866. An output of the summer 3835 provided to a second throw of the switch 3836. An output of the summer 3866 is provided to a second throw of the switch 3836.

Please amend the paragraphs beginning at page 68 line 19 as follows.

As shown in Figure 38, left and right stereo input signals are provided to left and right inputs ~~3803~~ 3801 and 3802 respectively. For the bass enhancement portion of the processing (corresponding to the bass enhancement block 101 shown in Figure 1), the left and right channels are added together by the summer 3808, processed as a monophonic signal, then added back into left and right channels by the summers 3828 and 3829 to form an enhanced stereo signal. The bass information is processed as a monophonic signal because there is typically little stereo separation in a bass frequency signal, so there is little need to duplicate the processing for the two channels.

Figure 38 shows software user controls including: a software control 3827 to control the amount of bass enhancement, a software control 3846 to control the width of the apparent sound stage, as well as software switches 3805, ~~3260~~3860, and 3836 to individually enable or disable the vertical, bass, and width image enhancements respectively. Depending on the application, these user controls can be either dynamically changeable or fixed to a specific configuration. The user controls can be "connected" to controls such as sliders, check boxes, and the like, in dialog box to allow the user to control the operation of the acoustic correction system.

Please amend the paragraph beginning at page 69 line 1 as follows.

After the elevation filters, the left and right channels are mixed together and routed through the low pass filter 3809 followed by the bank of bandpass filters 3810-~~3112~~3812. The low pass filter 3809 has a cutoff frequency of 284 Hz. Each of the following ~~four~~ three filters 3810-~~3112~~ 3812 is a second order band pass filter. The filter 3810 is selectable as either 40 Hz or 150 Hz. The filter 3811 is selectable as either`60 Hz or 200 Hz. Thus, there are three useful configurations for speaker size: small, medium and large. All three configurations use the three band pass filters, but with different center frequencies for the filters 3810 and 3811.